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Delta modulation for sound-signal distribution: A general survey

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RESEARCH DEPARTMENT

DELTA MODULATION FOR SOUND-SIGNAL DISTRIBUTION: A GENERAL SURVEY

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DELTA MODULATION FOR SOUND-SIGNAL DISTRIBUTION: A GENERAL SURVEY

Summary

Delta modulation has been proposed as an alternative to pulse code modulation (PCM) for sound-signal distribution.

This report describes the theory of operation of delta modulation and some of the modifications to the basic system which can be used to improve the overall performance or to optimise particular parameters to suit the form of the input signal. A comparison is made between delta modulation and PCM, and certain cases are pointed out in which delta modulation might be used with advantage as an alternative to PCM within the BBC.

1. Introduction

Delta modulation (ΔM), also known as 1-digit code PCM or 1-digit differential PCM, is a way of transmitting information by means of uniform pulses. In conventional pulse code modulation systems an n-digit code is generated which contains information on the absolute magnitude of each sample. In delta modulation a single digit only is generated at each sampling time, indicating in which direction the signal amplitude has changed; the transmitted pulses therefore carry information corresponding to the derivative of the input signal. At the receiving terminal the pulses are integrated and filtered to obtain the original signal.

Delta modulation was first proposed in the French laboratories of the ITT organisation and was the subject of French patents in 1946 and 1948.[1,2] A paper by de Jager (1953)[3] adequately explained the theory of operation of ΔM and the system performance. Over the last ten years there have been many written contributions on the subject. These have covered mathematical analysis as well as several modified forms including compandors and other techniques for improving the system performance. Alternative forms of coding have also been suggested, which are generally classified as direct feedback coding, delta modulation as originally proposed being a particular case of this general classification.

Although much has been written about delta modulation, practical applications have been few and largely restricted to telemetry, telephones and low-quality military communication systems. More recently it has been proposed as the method of modulation for the sound channel of a Sound-in-Syncs[4] system and also for the video channel of the Bell Picturephone[5] system.

Delta modulation has the advantage over pulse code modulation of simpler codec instrumentation, less stringent requirements for filters and greater resistance to transmission channel errors; moreover it does not require word synchronisation. In exchange for these advantages there are restrictions on input signal parameters, while for equivalent noise performance in high-quality applications, ΔM requires a higher transmission bit rate than PCM.

In this report the principles of delta modulation are explained and the system parameters discussed in connection with the transmission of sound signals. Some of the modified forms are discussed and comparisons are drawn between ΔM and PCM. Although this report deals mainly with sound signals, a brief mention is made of the application of ΔM to television. The conclusions suggest where ΔM could be used with advantage in the BBC.

2. Theory of delta modulation

2.1. Operation of basic system

At each sampling instant a delta modulation coder transmits a pulse which indicates the direction in which the

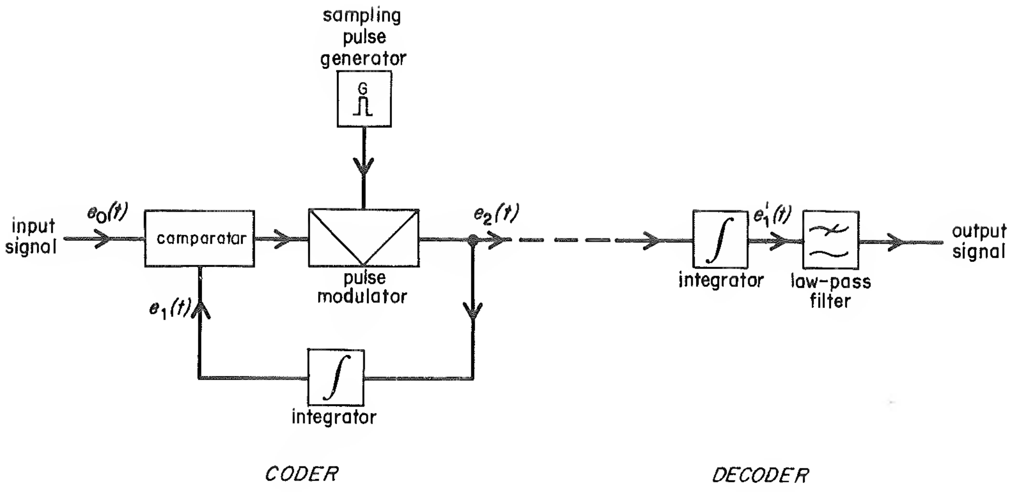


Fig. 1 - Basic circuit for ΔM

input signal has changed since the previous sample; a block diagram of a basic ΔM system is shown in Fig. 1.

The modulator operates on the sampling pulses in such a manner that positive pulses are delivered to line if the difference signal from the comparator is positive, and negative pulses if the difference is negative. The transmitted pulse train $e_2(t)$ therefore consists of positive or negative pulses at the same rate as the sampling pulses and there is no zero pulse state. The difference signal is obtained by comparing the input signal $e_0(t)$ with a signal $e_1(t)$ synthesised from the output pulses by integration. The comparator thus decides what polarity the output pulse should have in order to reduce the difference between the two voltages $e_0(t)$ and $e_1(t)$. The feedback signal $e_1(t)$ consists of a series of steps up or down, of amplitude δ, and is a 'quantised' approximation of the input signal.

In practice, the negative pulses are usually omitted from the transmission path to give a unipolar pulse train. The original pulse train can easily be reconstructed in the decoder by superimposing on the positive pulses a series of half-amplitude negative pulses at sampling frequency; positive pulses are thus reduced to half amplitude and zeros are replaced by the negative pulses.

In the decoder the received pulses $e_2(t)$ are integrated in a network similar to that in the coder to give a signal $e_1'(t)$ which is a close approximation of the input signal; error components due to the quantum nature of the pulses are present. Sampling components are removed in a low-pass filter. The difference between the original signal and the reconstructed approximation can be regarded as noise. This noise can be reduced by increasing the sampling pulse frequency so that, for the same peak signal capability, δ is smaller.

It will have been seen that for ΔM the transmitted bit rate is equal to the sampling rate, which must be several times greater than the highest frequency component of the input signal for satisfactory signal-to-noise ratio. In comparison, the sampling rate in PCM is approximately twice the highest input frequency but the transmitted bit rate will be n times the sampling rate where n is the number of digits in the PCM code; n also determines the signal-to-noise ratio.

2.2. Single integration ΔM

The simplest form of integrating network in the feedback path is a simple resistance-capacity combination R_1C_1 which has a large time constant. The response of this network to an impulse will approximate to a unit step. A typical transmitted pulse train and quantised feedback signal for a single integration system is shown in Fig. 2.

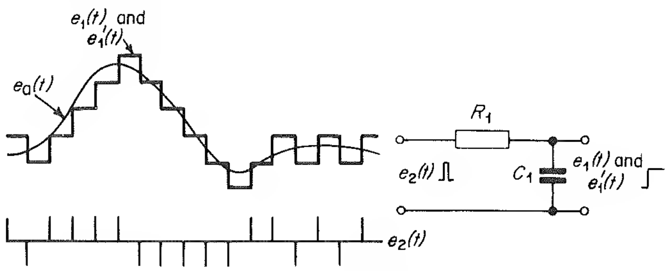


Fig. 2 - Single integration ΔM

2.3. Double integration ΔM

A much closer approximation to the original signal can be achieved by using a double integration network as in Fig. 3(a). The time constants R_1C_1 and R_2C_2 are both large and the response of the network to a unit impulse is a voltage which has a slope that either increases or decreases by a fixed amount with positive or negative input pulses. The feedback waveform is now a much smoother curve and is generally a much closer approximation to the input signal. A disadvantage of double integration however is that large changes in the slope of the input signal may not be recognised soon enough, since the comparator senses amplitude differences only, and large errors in the feedback signal may occur as illustrated in Fig. 3(a). Such errors are a form of overshoot which indicates that the system can tend towards instability. To overcome this disadvantage a further modification is made to the feedback network in the coder to

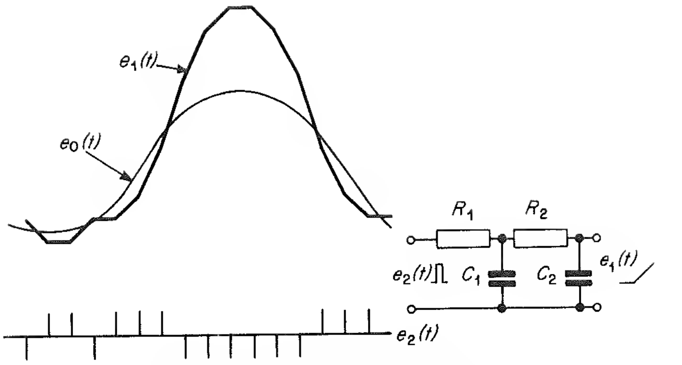


Fig. 3(a) - Double integration ΔM

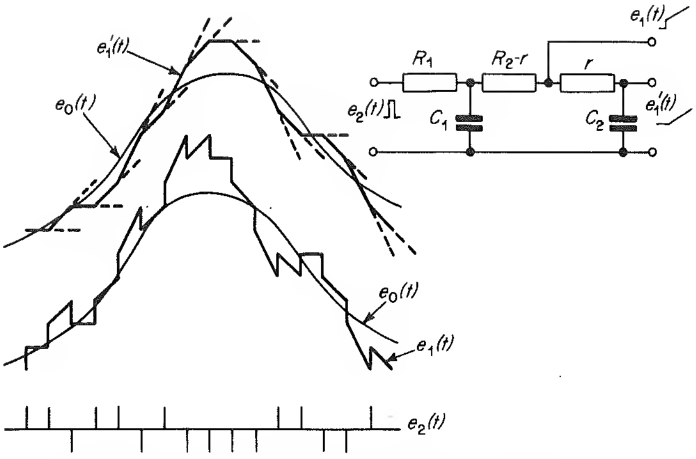


Fig. 3(b) - Double integration ΔM with prediction

afford a degree of prediction. The circuit of the integrator is now as shown in Fig. 3(*b*) and the response to a unit impulse $e_2(t)$ at the feedback point in the coder is a step followed by a voltage of constant slope. By virtue of the step, the output feedback signal $e_1(t)$ in the coder is a prediction of the level to which $e_1'(t)$, the voltage on C_2 in the decoder, will rise. This prediction is equivalent to extrapolation and the effect on the coding process can be seen from the curves of Fig. 3(*b*).

Let the prediction time $\tau = rC_2$ (1)

then

$$e_1(t) = e_1'(t) + rC_2 \frac{d}{dt} e_1'(t)$$

$$= e_1(t) + \tau \frac{d}{dt} e_1'(t)$$

$$= e_1'(t+\tau)$$

that is, $e_1(t)$ is the value that the voltage across C_2 will have at a time $(t+\tau)$.

If τ is too large the predicted signal deviates too far from the actual curve, and if τ is too small the system approaches that of normal double integration with its inherent instability. In addition, τ should be small compared to R_1C_1 and R_2C_2. The optimal value of τ is of the order of the time interval between sampling pulses.

Prediction is required only in the coder to obtain a closer approximation to the input signal, and in the decoder a conventional double integration network is used having time constants R_1C_1 and R_2C_2.

2.4. Threshold and overload

In the absence of an input signal, the output from a single-integration ΔM coder will consist of a series of alternating positive and negative pulses, and the feedback approximation signal will appear as in Fig. 4(*a*). This idling signal contains only high-frequency components and there will be no signal from the output of the low-pass filter in the decoder.

If the input signal has a peak-to-peak amplitude of less than δ, the alternating pulse sequence will be undisturbed and the output from the decoder will remain at zero. There is therefore a threshold below which information will not be transmitted.

For double integration the idling pattern is dependent on the degree of prediction, and can take either the form shown in Fig. 4(*a*) or a paired alternate pattern, i.e. ++−−++−−. Other patterns are possible but would indicate a value of τ not equal to the optimum value.

In considering the threshold condition as described above, it has been assumed that the contributions of +ve and −ve pulses to the integration process are equal; under this condition the coder is said to be balanced. If the contributions from the +ve and −ve pulses are not equal, an unbalance will exist in the integrator output which is observed as a disturbance in the idling pattern as shown in Fig. 4(*b*). The effect is analogous to a saw-tooth waveform added to the input signal and producing an unwanted signal in the decoder output. However, it was found experimentally that if the saw-tooth frequency is sufficiently far above the signal frequency band, then the wanted output is

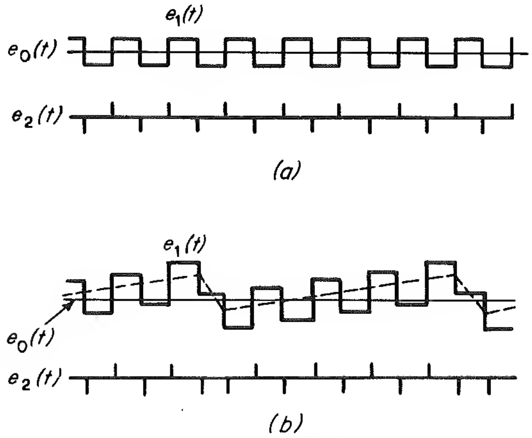


Fig. 4 - ΔM idling signals

(*a*) Balanced condition (*b*) Unbalanced condition

negligible. Under these conditions the effect of unbalance is equivalent to the addition of a perturbing or dither waveform which has the effect of reducing the threshold level. It has been found by several investigators that it is difficult to maintain threshold stability in ΔM systems. Feedback techniques can be employed to maintain a balanced condition but it is extremely difficult to hold a precise degree of unbalance. The tendency towards instability increases with sampling rate.[6]

In a ΔM system the information conveyed by the transmitted digits correlates with the first derivative of the input signal. There is therefore no fixed maximum amplitude of the input signal but the system will overload when the slope of the input signal exceeds a value defined by the system parameters; this condition is referred to as slope overload.

The largest slope that the system can transmit without error is one which changes by one level per sample. Thus for a sampling frequency of f_s samples per second and a step amplitude of δ volts the maximum slope will be δf_s volts per second.

The maximum slope for a sinusoidal signal of frequency f, and peak amplitude S_m volts is $S_m 2\pi f$ volts per second. In order to transmit this signal without distortion we have: —

$$S_m \leq \frac{f_s \delta}{2\pi f} \text{ volts} \qquad (2)$$

It follows that the maximum amplitude that can be transmitted without distortion is reduced by 6 dB/Octave as the frequency increases; the number of discrete levels in the integrated signal will also decrease.

Fig. 5(*a*) shows, as a function of frequency, the maximum signal amplitude that can be transmitted without distortion by a single integration ΔM system. The overload characteristic may be modified as shown in Fig. 5(*b*) for the case of double integration ΔM if the break frequency, $f_2 = 1/(2\pi R_2 C_2)$, of the second stage of the integrator is lower than the upper frequency limit, f_0, of the input signal; in this case the slope of the overload characteristic is increased at frequencies above f_2.

The characteristics of delta modulation systems are well matched to speech signals which have a power spectrum that falls off with increasing frequency. However, for music, which can have large amplitude high-frequency components, ΔM may introduce distortion through slope overload.

2.5. Quantising noise

Quantising noise is defined as the error between the coder input signal and the filtered decoder output signal. In ΔM the quantising noise has a random amplitude which is dependent on the input signal level. This can be fairly easily demonstrated by considering the nature of the error at three signal levels. At zero input signal there is no noise component in the filtered output signal. For signals between threshold and overload the output signal is delayed by one clock period with respect to the input signal, due to

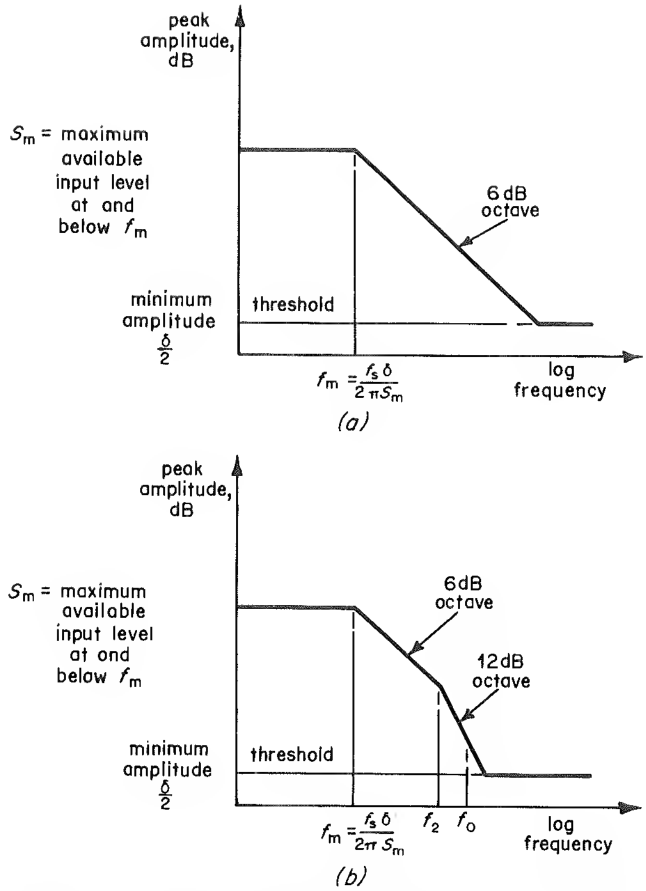


Fig. 5 - Maximum amplitude that can be transmitted without distortion for ΔM

(*a*) Single integration (*b*) Double integration

the code-decode process; the error is therefore dependent on signal frequency and amplitude. Above overload the output delay increases and the error will depend on the degree of overload.

Several contributions have been made towards analysis of ΔM noise,[3,7,8,9,12] the simplest presentation being for the case of a sine wave near overload.

It is usual to assume[7] that quantising noise is of random magnitude, having however a spectral periodicity at sampling frequency f_s; it is also assumed that the frequency distribution of the noise power is of the form $(\sin x/x)^2$ with the first null at f_s. The total power of this distribution equals the power that results if the level at the origin is held constant over a band $f_s/2$. Tomozaura and Kaneko[12] give a formula:

$$\frac{S_m}{N} = \frac{\pi}{16\alpha}\left(\frac{f_s}{f_0}\right)^{1/2} \left| \frac{H(f)}{H(f_s/2)} \right| \qquad (3)$$

for the ratio of the maximum signal level S_m at the overload point to the average quantising noise level N at a given modulation frequency f. In Equation (3), as before, f_s is the low-pass bandwidth; $H(f)$ is the transfer function of the decoder and α is the ratio of the r.m.s. quantising error to the quantising step. It was found[12] that

$$\left.\begin{array}{l}\alpha = 0{\cdot}376 \text{ for single integration} \\ \alpha = 0{\cdot}668 \text{ for double integration}\end{array}\right\} \qquad (4)$$

For single integration, $H(f)$ must be replaced by $H_1(f)$

$$H_1(f) = \frac{1}{1 + j(f/f_1)} \tag{5}$$

$f_1 = 1/(2\pi R_1 C_1)$ and $f_1 \ll f \ll f_s$. It follows that the last factor on the right hand side of Equation (3) can be taken as $(f_1/f) \div (2f_1/f_s)$ or $(f_s/2f)$, and therefore Equation (3) reduces to:

$$\begin{aligned}\frac{S_m}{N_1} &= \frac{\pi}{32 \times 0{\cdot}376 f} \cdot \frac{f_s^{3/2}}{f_0^{1/2}} \\ &= 0{\cdot}261 \cdot \frac{f_s^{3/2}}{f \cdot f_0^{1/2}}\end{aligned} \tag{6}$$

This agrees with de Jager's formula[3] for single integration, except insofar as de Jager's numerical coefficient is 0·20.

For double integration, on the other hand, $H(f)$ must be replaced by $H_2(f)$ where:

$$H_2(f) = \frac{1 + j(f/f_3)}{[1 + j(f/f_2)]\ [1 + j(f/f_1)]} \tag{7}$$

$$f_3 = 1/(2\pi r C_2), f_2 = 1/(2\pi R_2 C_2) \text{ and } f_1 \ll f \ll f_3;$$

in practice, f_2 is made equal to f_0.

Whatever the relative values of f, f_1, f_2, f_3 and f_s, a formula for the double integration signal-to-noise ratio, S_m/N_2, can be obtained from Equation (3) by substituting $H_2(f)$, and $H_2(f_s/2)$ from Equation (7). The value of α is now 0·668, and it is again assumed that for stability $f_s = 2\pi f_3$, i.e. $f_s \gg f_1$.

Then:—
$$\frac{S_m}{N_2} = 0{\cdot}0223 \cdot \frac{f_s^{3/2}}{f \cdot f_0^{1/2}} \cdot \left[\frac{4f_2^2 + f_s^2}{f_2^2 + f^2}\right]^{1/2} \tag{8}$$

By applying the further condition of $f \ll f_2 = f_0$ we have:—

$$\frac{S_m}{N_2} = 0{\cdot}0223 \cdot \frac{f_s^{5/2}}{f \cdot f_0^{3/2}} \tag{9}$$

which agrees with de Jager's formula[3] with the exception of the numerical constant which de Jager quotes as 0·026.

Similarly, formulae for signal-to-noise ratio quoted in other references[7,9,11,13] can be obtained from Equation (8) by applying the appropriate assumptions.

The Equations (6) and (9) are the forms that are usually quoted, and it can be seen that quantising noise varies with sampling frequency according to a three-halves power law for single integration and a five-halves law for double integration. For both systems the signal-to-noise ratios are inversely proportional to signal frequency.

3. System parameters

For the present purpose theoretical and practical examinations were made of delta modulation parameters and performance. Two applications were considered, one a system for the transmission of high-quality sound signals, the other, a lower quality system for the transmission of speech.

3.1. ΔM for transmission of high-quality sound-signals

From Equations (6) and (9) in Section 2, but substituting the coefficients given by de Jager, the curves in Fig. 6 have been constructed to show the variation of signal-to-noise ratio with bit rate. For a ΔM system the parameters have been chosen as 14 kHz for the bandwidth and 1 kHz for the signal frequency. These figures are representative of the bandwidth and 'mean' frequency of a high-quality system. In a practical system 1 kHz would probably be chosen as the highest frequency that could be transmitted at peak level. For comparison the equivalent curve for a PCM system is shown for a 32 kHz sampling rate and the number of digits per sample as indicated. This latter curve is applicable for all signal frequencies in a 14 kHz bandwidth.

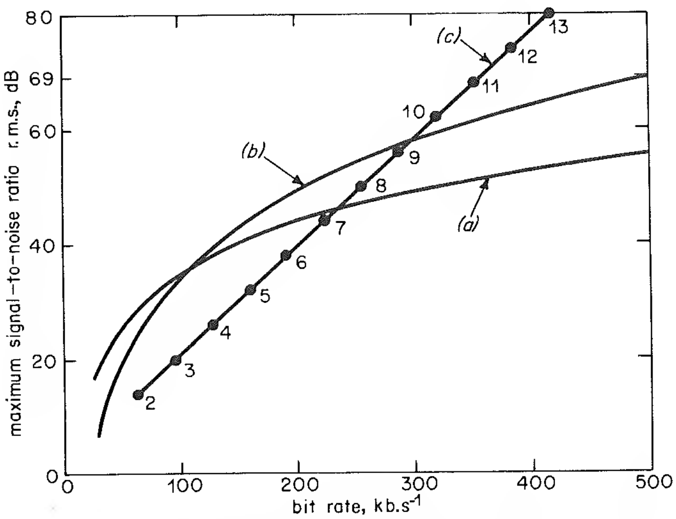


Fig. 6 - Variation of signal-to-noise ratio with bit rate:— wide band audio system

(a) Single integration ΔM f = 1 kHz, f_0 = 14 kHz
(b) Double integration ΔM f = 1 kHz, f_0 = 14 kHz
(c) PCM f_s = 32 kHz No. of digits as shown

The acceptance figure for the ratio of r.m.s. signal*-to-r.m.s. unweighted white noise of a high-quality sound-signal system is 69 dB.[10] Assuming for simplicity that the system comprises nothing more than one coder and decoder, Fig. 6 shows that this level can only be obtained with double integration ΔM at a bit rate of 500 kb.s^{-1} and at a considerably higher rate for single integration ΔM. In comparison, the acceptance figure can be obtained with a straight 11-digit PCM system requiring a bit rate of 352 kb.s^{-1}.

As with PCM, the noise performance of a ΔM system can be improved by employing companding techniques, and such a system has been proposed[4] for a high-quality sound channel employing a bit rate of 256 kb.s^{-1}. For this and lower bit rates straight ΔM shows a noise advantage over PCM.

* By convention, the input signal is assumed to be a sinusoid which fully loads the system.

For the present investigation, it was decided to construct a 14 kHz bandwidth ΔM system based on a bit rate of 256 kb.s^{-1}, the resulting signal-to-noise ratios being such as would allow simple measurements and comparisons with PCM, and also offer a system for testing various companding techniques.

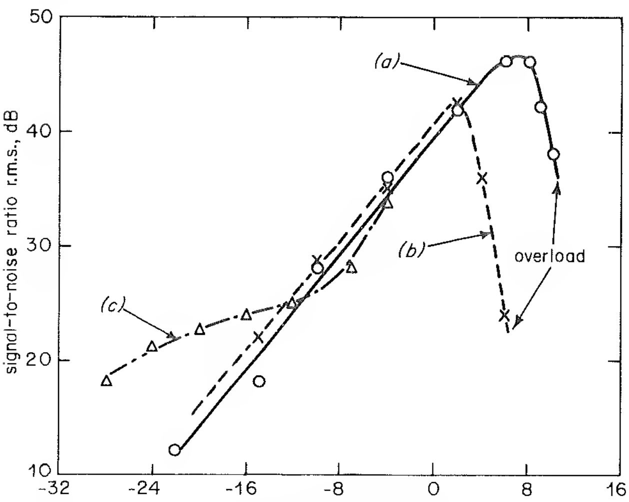


Fig. 7 - Input level/noise characteristic signal integration ΔM

f_s = 256 kHz f_0 = 14 kHz

(*a*) f = 1 kHz (*b*) f = 2 kHz (*c*) f = 1 kHz, unbalanced coder

The experimental coder was adjusted to handle peak signals of +8 dBm up to a signal frequency of 1 kHz. The measured variation of signal-to-noise ratio with input level for single integration ΔM is shown in Fig. 7 which also illustrates the 6 dB change in overload point when the input frequency is raised by one octave from 1 kHz to 2 kHz. Also shown in this figure is the effect of an unbalanced coder with the fundamental of the sawtooth components at 28 kHz; the signal-to-noise ratio at low signal levels is increased by the dither, although as pointed out previously, this condition of unbalance is difficult to maintain accurately.

Fig. 8 shows the increase in signal-to-noise ratio obtained by double integration. Curve (*b*) is for the preferred case where the second integrator break frequency f_2 is at f_0 – in this case 14 kHz – and the overload characteristic has a 6 dB/Octave slope; for curve (*c*), f_2 has been reduced to 1 kHz resulting in a further increase in signal-to-noise ratio, obtained however at the cost of an overload characteristic falling at 12 dB/Octave.

It was observed during listening tests that for double integration ΔM the quantising noise was not only at a lower level but differed in character from that obtained with single integration, especially in the reproduction of low-level signals near threshold. For single integration the noise has the characteristics of a line spectrum, whereas the double integration noise has a smoother spectrum and is not dissimilar to PCM quantising noise. For the chosen system parameters, the experimental double-integration ΔM system was judged to be equivalent to a 9-digit PCM system, a

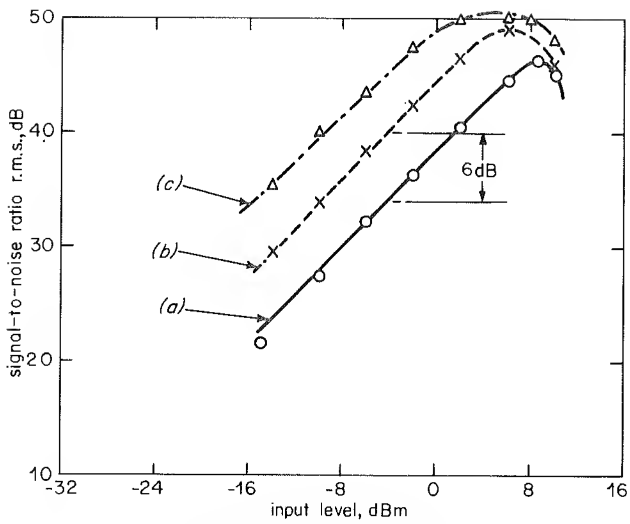


Fig. 8 - Input level/noise characteristics of ΔM with 1 kHz input signal

f_s = 256 kHz f_0 = 14 kHz

(*a*) Single integration

(*b*) Double integration, f_2 = 14 kHz, overload 6 dB/Octave

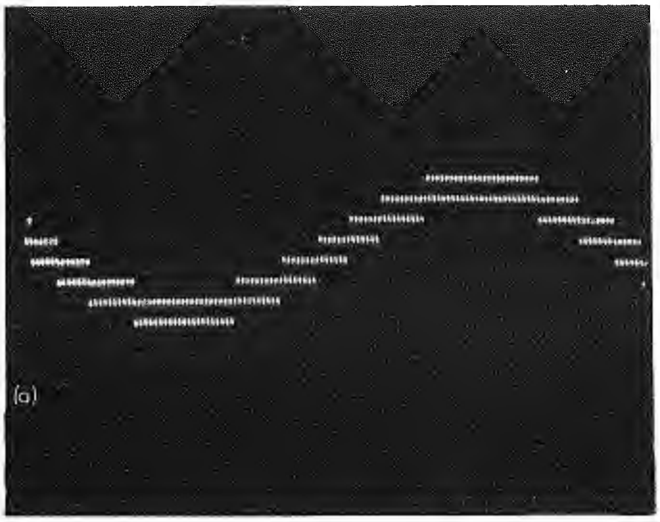
(*c*) Double integration, f_2 = 1 kHz, overload 12 dB/Octave

result which is in good agreement with theoretical prediction. The photographs of Fig. 9 show the feedback waveforms of single and double integration for a 1 kHz signal 10 dB below the overload point, and illustrate the smoother approximation of double integration to an original sine wave input.

During overload in a ΔM system an input sine wave is transformed to a triangular wave, the slope of which is equal to the maximum slope of the system. Distortion during overload therefore takes the form of the introduction of high-order harmonics, which are subjectively very annoying. High-quality sound-signals can contain high-level high-frequency components which can cause overloading and the subjective effect of momentary distortion can be very disturbing.

3.2. ΔM for the transmission of telephone-quality speech

The spectral distribution of average speech falls off at approximately 6 dB/Octave above a frequency of about 800 Hz. Delta modulation is therefore well suited as a transmission system for speech. Fig. 10 shows the variation of signal-to-noise ratio with bit rate for single- and double-integration ΔM having a 3·5 kHz bandwidth, and for comparison purposes, PCM with an 8 kHz sampling frequency. A crossover exists at a bit rate of approximately 40 kb.s^{-1} below which ΔM offers advantages over PCM in signal-to-noise ratio. At these bit rates the signal-to-noise ratio is low and applications of ΔM are therefore mainly restricted to communication systems where it is required to transmit intelligible speech over a low-capacity channel. A simple subjective test indicated that speech would be understood without significant concentration on the part of the listener when the signal-to-noise ratio is 22 dB. This performance can be obtained with ΔM operating at a bit rate of 22·5 kb.s^{-1} provided that the signal fully loads the system.



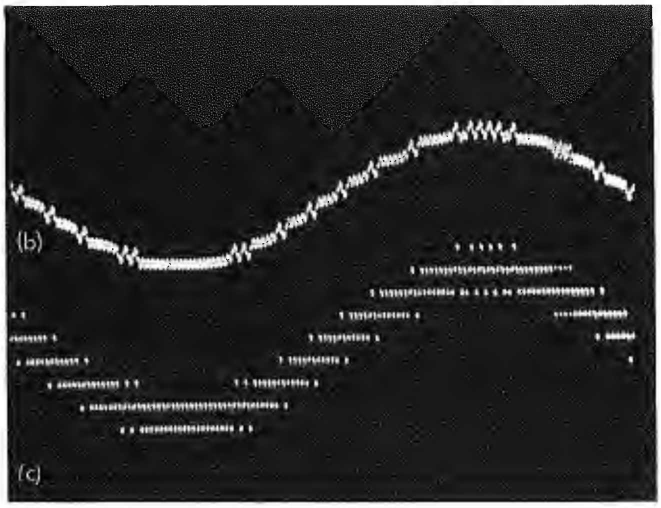


Fig. 9 - ΔM coder waveform: f_s = 256 kHz, f = 1 kHz at 10 dB below overload

(*a*) Feedback signal: single integration
(*b*) Feedback signal: double integration
(*c*) Output of first integrator: double integration

The dynamic range of the system would be low, but could be increased by using compandors or other noise reduction techniques. Several methods of companding have been proposed and these will be discussed in the following section.

4. Noise reduction techniques

Noise reduction techniques used in digital communication systems operate on the principle of compressing the dynamic range of the input signals so that the quantising step remains small compared to the signal, or alternatively, of graduating the size of the quantising steps. In the decoder a complementary expansion takes place to restore the signals to their original level, with a corresponding reduction in quantising noise. The noise reduction is beneficial mainly for low-level signals since high-level signals usually mask the quantising noise.

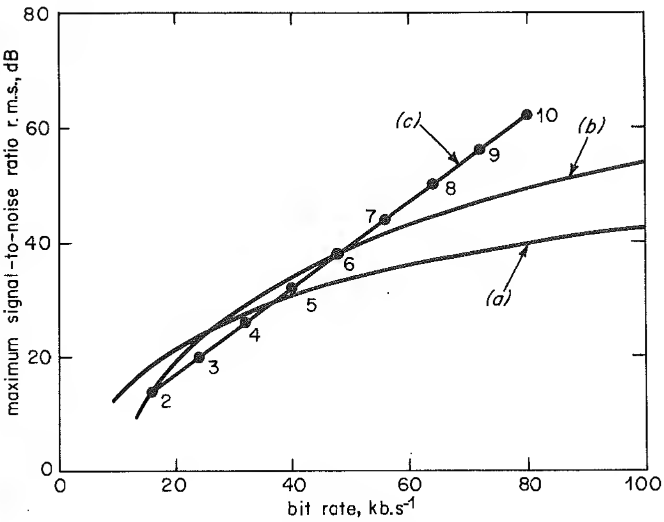


Fig. 10 - Variation of signal-to-noise ratio with bit rate:- narrow band audio system

(*a*) Single integration ΔM, f = 800 Hz, f_0 = 3·5 kHz
(*b*) Double integration ΔM, f = 800 Hz, f_0 = 3·5 kHz
(*c*) PCM f_s = 8 kHz, number of digits as shown

There are two forms of compandor; instantaneous compandors which employ some fixed non-linear device, or graduated quantising steps, and syllabic compandors in which gain is varied in accordance with the level of the input signal but is substantially constant over a number of cycles of the signal waveform. In the latter case it is often necessary to convey the companding information to the decoder, preferably by the main digit stream, or by an additional communication channel if sufficient capacity is available.

In all cases the aim of companding is to modify the signal-to-noise/input level characteristic to a form as shown in Fig. 11.

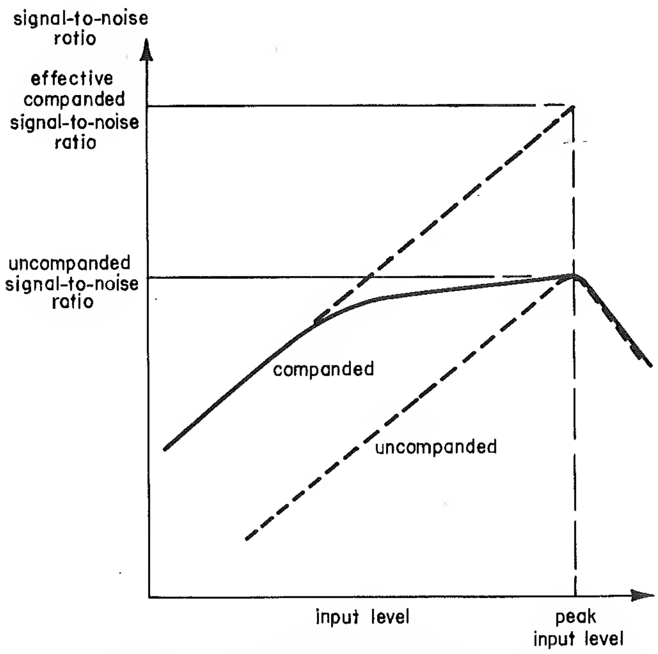


Fig. 11 - Effect of companding on signal to noise ratio/ level characteristics

Pre- and de-emphasis as used for noise reduction in analogue modulation systems and PCM cannot be usefully applied to ΔM since the pre-emphasis at the coder will cause premature overloading at high frequencies.

4.1. Instantaneous compandors

Instantaneous compandors are usually restricted to low-quality systems for the following reasons. Waveform distortion can occur if the transfer characteristics of non-linear devices used for compression and expansion are not exactly complementary; errors in matching are more likely if a large range of companding is required. Distortion can also arise in systems using graduated quantising steps due to the inability to reproduce accurately low-amplitude information that may be supimposed upon large-amplitude components of the input signal. With graduated quantising there is no difficulty in making the compression and expansion laws accurately complementary.

4.2. Syllabic compandors

Syllabic compandors as devised for ΔM operate mainly in the feedback network and fulfil the dual role of noise reduction and the reduction of slope overload distortion. All the methods to be described function by changing the quantising steps in the feedback loop.

Fig. 12 illustrates two basic forms of syllabic compandor; the first derives its control from the input signal and requires a separate channel, which can be either analogue or digital, to convey the control to the decoder.[11] The second derives the control from the transmitted digits and therefore uses the digit stream as carrier for the control information.[4,12,13,14] Full details of the various proposed systems will not be described here since this information can be obtained from the quoted reference, and it will be sufficient to outline the suggested methods of level detection and control.

Level detection and generation of the control signal may be realised as follows:—

(a) By differentiation, rectification and filtering of the input signal. The differentiation is included to obtain information on the slope of the input signal, and the filter determines the control time constants. A system using this technique[11] transmits the control signal to the decoder by a secondary ΔM coder operating at approximately one sixteenth the bit rate of the main coder, the output pulses being multiplexed into the main digit stream.

(b) By using a secondary integrating decoder network on the output digit stream, followed by a peak detector.[12] The time constant of the additional inte-

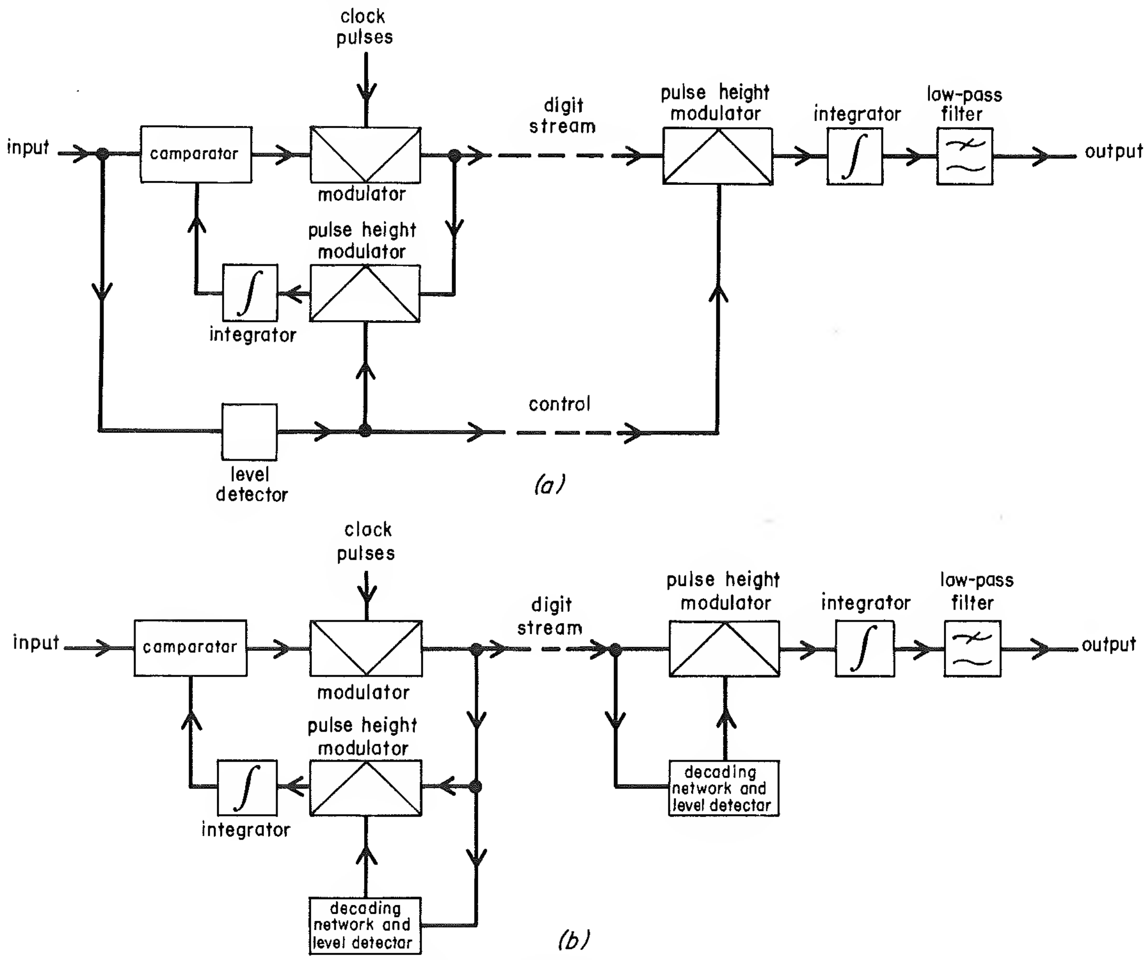


Fig. 12 - Companded ΔM systems

(*a*) Companded ΔM system with separate control chain (*b*) Companded ΔM system with integral control chain

grating network will determine the rate of control. This technique operates indirectly on the amplitude of the input signal, and a threshold must be included in the detector so that the compandor operates only over a predetermined dynamic range.

(c) By extracting the derivative of the input signal by means of a low-pass filter on the digit stream.[13] The filter bandwidth will determine the control time constants. In a system adopting this technique the control signal could be transmitted to the decoder as a low-frequency signal component below the nominal low-frequency cut-off of the input signal. The control signal is then simply extracted at the decoder with a L.P. filter. In such a system however the rate-of-change of the control signal would be low.

(d) By examining the digit stream in logic networks to detect successive strings of 1's or 0's;[14] these digit states indicate slope overload. Separate smoothing of the logic output is needed to obtain the required time constant of the control signal.

Control of the coder and decoder quantising step sizes may be realsed as follows:

(a) By using the control signal to vary the height of the digit pulses feedback into the integrating network.[11,12,13] Pulse height modulators performing this function can be very simple, for example, a transistor, operating near cut-off, whose bias is varied by the control signal.

(b) By changing the quantising step size according to a predetermined pattern.[14] For example if the transmitted digit stream contained a series of 1's (or 0's) the step size would be progressively increased in say the ratios 1:2:4:8: etc. At the end of the series of like digits the step size would either immediately return to unit value, or return through the intermediate ratios. There are many forms which adaptive systems like this can take and the ratios can be individually optimised to suite the form of the input signal.

4.3. Compandor performance

The syllabic companding systems as described have been mainly applied to telephone systems, where the inherent low quality and narrow bandwidths have masked some of the undesirable side effects of companding. In these circumstances degrees of companding up to 26 dB have been achieved thus making the signal-to-noise performance of low-bit ΔM systems quite acceptable for the transmission of intelligible speech.

Syllabic companding has also been proposed for a wideband music system.[4] The degree of companding suggested gives an effective r.m.s. signal-to-r.m.s.-noise ratio. for a 14 kHz bandwidth, 256 kb.s^{-1} system, of approximately 62 dB.

To evaluate the performance of a high-quality companded system, the experimental laboratory equipment referred to in Section 3 was modified to include a secondary integrator and pulse height modulators. The modifications gave approximately 8 dB increase in the signal-to-noise ratio and some protection against slope overload.

The instrumentation of this experimental arrangement was not ideal and difficulty was experienced in maintaining an accurate coder balance, but it was possible to carry out some rough listening tests. Although the average noise level was reduced by the action of the compandor there was a slight increase in the programme-modulated noise component over that which normally occurs with straight ΔM. With the amount of companding available it was difficult to detect any change in the amount of overload distortion.

Although the tests were by no means comprehensive it was felt that the shortcomings of the system were too great for any further companding to render the system satisfactory for the transmission of high-quality music.

4.4. Slope overload protection

A novel scheme has been recently proposed[15] to reduce the distortion due to slope overload. A delay line of time delay T is included at the input to the coder, so that the difference in level, y, between the input and output of the delay line gives an indication of the slope of the input signal y/T. When this value exceeds the maximum slope for the coder appropriate modifications are made to the coder parameters.

5. Direct feedback coding

Direct feedback coding, so called since no processing takes place in the feedback loop, is a refinement of delta modulation; in fact, the latter could be regarded as a particular case of single-bit direct feedback coding. There are many forms of direct feedback coding, each one having its parameters optimised for a particular input signal. The most well-known form is delta-sigma modulation (ΔΣM)[16] and the principles of operation can best be examined by considering modifications to conventional ΔM.

5.1. Delta-sigma modulation

It has been shown that conventional ΔM has the disadvantage of the dynamic range and signal-to-noise ratio being inversely proportional to the signal frequency. This disadvantage arises through the inevitable differentiation of the input signal, but it can be overcome if an integration process precedes the coder input with complementary differentiation following in the decoder. A ΔM system thus modified is illustrated in Fig. 13(*a*). The input to the pulse modulator is the difference between the input signal $e_0(t)$ and the output signal $e_2(t)$ after each has been integrated. The transfer characteristics of the integrators are identical and the two networks can therefore be replaced by a single integrator following the comparator. At the decoder the integrator and differentiator are exactly complementary and can therefore be omitted. The modified system now appears as in Fig. 13(*b*) which is a basic delta-sigma modulator.

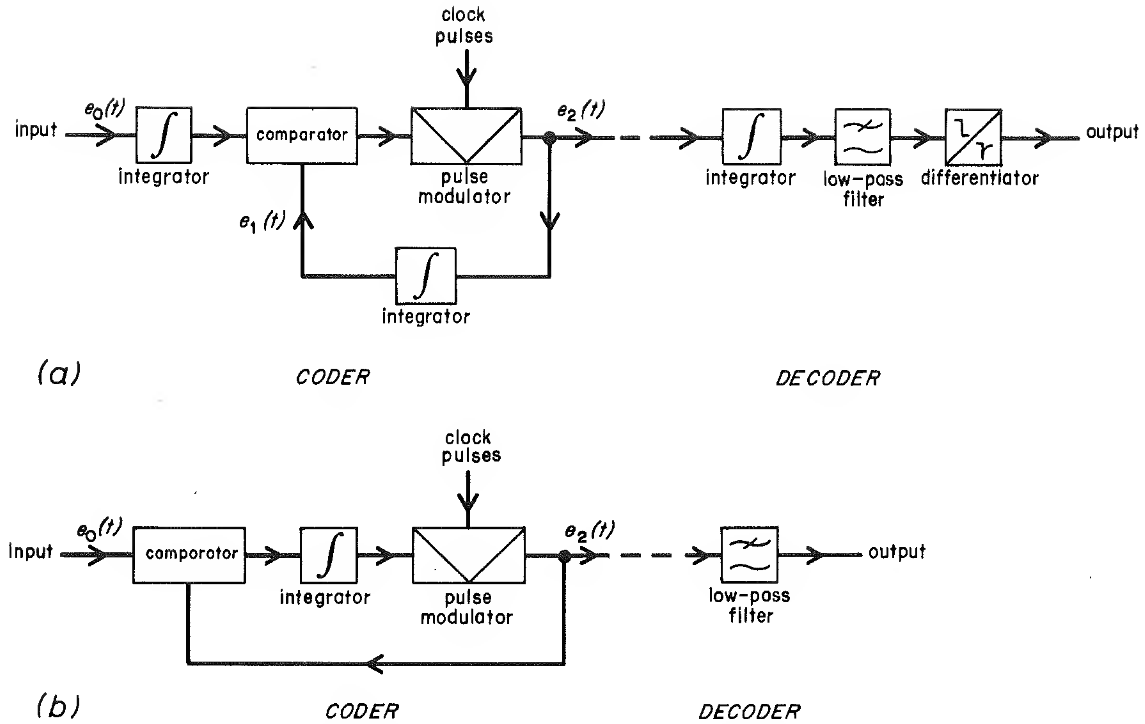


Fig. 13 - Derivation of ΔΣ modulation for ΔM
(*a*) Modified ΔM system (*b*) ΔΣ system

The advantage of ΔΣM is that the system has a flat amplitude/frequency characteristic and overload occurs at the same level for all frequencies. In exchange for a flat frequency characteristic the spectrum of the quantising noise is no longer flat but rises at 6 dB/Octave and the coding threshold is similarly modified.

By a process similar to that already illustrated in Section 2, the r.m.s. signal to quantising noise ratio for maximum signal amplitude can be shown to be:

$$K_3\left(\frac{f_s}{f_0}\right)^{3/2} \tag{12}$$

where K_3 is a constant which has been computed[7] to be $3/(4\pi)$.

The output pulses carry information corresponding to the amplitude of the input signal, and as the input signal increases in amplitude the output pulses appear more frequently. Demodulation of the pulses at the decoder is performed with a low-pass filter only.

Syllabic companding can be applied to ΔΣM in a similar manner to that previously described for ΔM. A control signal can be developed by logical examination of the digit stream[17] (a series of 1's or 0's will represent maximum amplitude), or the input signal can be reconstructed by filtering and a control signal generated by peak detection.

5.2. Generalised form of direct feedback coders

The integrator preceding the pulse modulator in the ΔΣM coder may be a single or double integrator, or more generally, any signal processing network that modifies the quantising noise spectrum to suit a particular system performance or form of input signal. Similarly, in addition to shaping the noise spectrum, it is possible to modify the signal spectrum by adding appropriate pre- or de-emphasis networks before the coder with the complementary networks at the decoder.

Direct feedback coders can therefore be tailored to suit the parameters of the signal to be transmitted.[18]

6. Comparison between PCM and ΔM

In making a comparison between PCM and ΔM it is clear that the latter has advantages in the form of simpler instrumentation, less stringent requirements for filters, and with its 1-bit code does not require word synchronisation.

When comparing the ability of the two systems to transmit a signal it has to be remembered that the performance depends on the nature of the input. For high-quality low-noise applications ΔM requires a larger transmission bandwidth than PCM, whereas the converse is true for low-quality systems in which the presence of noise is not a serious impairment as long as the message is intelligible. The maximum amplitude/frequency characteristic and noise spectrum for PCM are flat, whereas ΔM has a flat noise spectrum but a 6 dB/Octave roll-off in the maximum amplitude/frequency characteristic. In the latter case, however, the system parameters can easily be modified to suit the form of the input signal. This can be advantageous for example in the case of speech which does not have a flat spectral distribution.

Further comparisons between PCM and ΔM have been made mathematically from the point of view of information

theory.[7,9] Both systems have nearly the same capacity for transmitting information for an equal number of quantising levels, provided the number of levels is larger than 10. However, as already stated, for high signal-to-noise ratios ΔM needs a larger bandwidth.

In PCM, successive samples can take any of the 2^n possible quantised levels, whereas in ΔM the quantised signal can only change by one positive or negative quantising step δ at each sample. This means that the number of pulses to be transmitted for a given change of input signal is on average less for ΔM than for PCM and it follows that less signal power is required for ΔM.

The lower power requirements of ΔM imply that the system is more resistant to transmission channel errors than PCM. Certainly the end effect of digit errors in the two systems is different. In PCM, the error in the output will depend on which digit within a group is wrong. If a most significant digit is wrong, the error in the output can be half the maximum peak-to-peak signal level, whereas if a least significant digit is wrong, the error in the output is equal to the level of a quantising step. In ΔM the effect of digit errors is always the same, that is an output error equal to twice the quantising step. However, in the event of a burst of digit errors the error in the decoder output can be large, due to the cumulative effect of the integration process. It is of interest to note that with a delta-sigma system, which does not have a decoding integrator, a greater number of digit errors can be tolerated than with ΔM.

Following a break in a transmission link, PCM will return to the correct absolute level from the decoder as soon as word synchronisation is restored. In ΔM no word synchronisation is necessary but the absolute level will not be correct until the integration process has re-established the mean level.

7. Conclusions

This survey has described the fundamentals of delta modulation and some of its variants. The system parameters and performance have been examined and compared with pulse-code modulation.

Delta modulation has the advantage over PCM of relative simplicity and low cost of the coding equipment, but in exchange puts restrictions on input signal parameters and for high-quality applications requires a higher transmission bandwidth for the same signal-to-noise ratios.

A broadcast authority is required to maintain a high quality yet must have due regard to the efficiency with which the high standards can be achieved. To this end, delta modulation offers no advantages over PCM for the transmission of high-quality sound. However, some low-grade internal communications are often required within a broadcasting network for the purposes of administration and control. These communications may take the form of data, speech or possibly facsimile, and it is here that delta modulation can offer advantages over PCM by reason of lower transmission bandwidth and reduced cost of terminal equipment.

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